

1 **METHOD OF IMPROVING AUDIO PERFORMANCE AND**
2 **POWER UTILIZATION OF A PORTABLE AUDIO DEVICE WITH**
3 **ELECTRONIC ANTI-SHOCK SYSTEM (EASS)**
4 **BACKGROUND OF THE INVENTION**

5 1. Field of the Invention

6 The present invention is related to a method of improving performance
7 and power utilization of a portable audio device fitted with an electronic anti-
8 shock system (EASS), and more particularly to a method of improving
9 performance and power utilization of a portable audio device such as CD player
10 by employing MPEG, a high compression rate algorithm, to encode/decode
11 audio signals so as to increase the capacity of audio data stored in a temporary
12 memory, as a result extending the buffering time, and realizing power saving.

13 2. Description of Related Art

14 CD players usually have an electronic anti-shock system (EASS) or
15 equivalent buffering device that creates a data buffer in the signal processing
16 path between the data retrieval lens and the audio signal processor to prevent
17 interruptions in audio playback. If CD players are subjected to shocks or
18 vibrations during reading of audio data, audio signal processing will be
19 interrupted, and the quality of audio output will be degraded accordingly.

20 A conventional EASS has the function of a CODEC, a coding-decoding
21 device that converts audio signals into digital bit streams and back again, and the
22 basic structure is shown in Fig. 3, in which separate data paths are used to
23 process audio data going through the right and left audio channels. The input
24 pulse code modulation (PCM) signals going through the right and left channels

1 are respectively processed by a pair of ADPCM encoders (71) (71'), and then
2 saved in a pair of memory devices (DRAM) (72) (72'). For faithful reproduction
3 of the sound, audio compressed data is read from the above memory devices
4 (DRAM) (72) (72'), and then passed to a pair of corresponding ADPCM
5 decoders (73) (73') to restore to the original PCM signals for the right and left
6 channels. Thereafter, the decoded audio signals in the right and left channels are
7 simultaneously passed to an audio signal processor (74) for output through a
8 speaker. Basing on the above structure, the above mentioned temporary memory
9 in the audio processing path has a buffering effect that can somewhat prevent
10 vibration-caused interruptions during audio playback.

11 However, the conventional electronic anti-shock system (EASS)
12 employs the adaptive differential pulse code modulation (ADPCM) for encoding
13 and decoding, by which the waveform of the analog signals are sampled at a
14 fixed frequency.

15 Generally, audio data have to be compressed before they can be saved in
16 a memory. The relationship between data size and the compression rate will be
17 explained hereunder, as it concerns the utilization of memory resources. For
18 example, the waveform of 12 bit/sample is compressed with a 3:1 ratio to
19 become 4 bit/sample, and when reading out data from memory, the data is
20 decompressed with 1:3 ratio to restore to the original audio format of 12
21 bit/sample for sound reproduction. Therefore, the memory used to save the audio
22 data is less than that with standard PCM codes, but the compression rate is not
23 adequate for saving large amounts of audio data in the limited memory of a CD
24 player.

1 The compression coding scheme for the above-mentioned conventional
2 ADPCM encoder (71) can be implemented in either 3-bit mode or 4-bit mode.
3 Since all current audio equipment has at least two audio channels, the 4-bit
4 operation mode is selected in this example for calculation of the bit rate with a
5 sampling frequency of 44.1 KHz:

6 $4(\text{bits}) \times 44100 \times 2 (\text{number of audio channels}) = 352,800 \text{ Kbps}$

7 If the operation is in 3-bit operation mode, the bit rate is:

8 $3(\text{bits}) \times 44100 \times 2 (\text{number of audio channels}) = 264,600 \text{ Kbps}$

9 If the memory installed in the above anti-shock system is 16M bits, as in
10 the present example, the required buffering time for the two operation modes can
11 be:

12 4-bit mode: $16,000,000 \div 352,800 = 45.35 \text{ (sec)}$

13 3-bit mode: $16,000,000 \div 264,600 = 60.46 \text{ (sec)}$

14 From the above explanation, the buffering time of the EASS is
15 dependent on the bit rate and the memory capacity. The data saving operation in
16 the EASS has to be sustained for a longer duration if the performance of a CD
17 player having EASS is to show any noticeable improvement.

18 Within the constraint not to increase a DRAM memory, the only way to
19 improve the performance of CD player is to increase the bit rate in the audio
20 compression. If the compression rate is increased, the performance of EASS can
21 be improved without using additional memory.

22 However, the ADPCM algorithm primarily is not primarily designed for
23 audio data compression, as the above compression rate cannot support high
24 capacity storage in limited memory storage of a portable CD player. Satisfactory

1 buffering will need large amount of DRAM memory, which is quite difficult for
2 a compact sized CD player, not to mention the increased costs.

3 From the foregoing, it is quite clear that ADPCM cannot satisfy the
4 present requirement of data conversion with high compression, but there are
5 some more advanced compression algorithms, such as the MPEG layer I and II,
6 which are able to produce reasonably acceptable sound quality with much better
7 utilization of memory than ADPCM. Audio data with double or triple larger size
8 can be saved in the same amount of memory space as compared with the
9 conventional ADPCM-based systems.

10 For portable CD players, another benefit of using MPEG in the EASS is
11 the power saving feature. When the audio data is read from the data buffer, the
12 servomotor of the CD player is kept in a suspended mode. Therefore, the longer
13 the CD servo can be suspended by the operation in the data buffer, the less the
14 system power is used.

15 Therefore, the present invention attempts to incorporate an audio
16 compression algorithm having high compression rate in the EASS to attain the
17 most desirable balance point between audio performance, power management,
18 and costs.

19 **SUMMARY OF THE INVENTION**

20 The main object of the present invention is to provide a method for
21 improving the audio performance of a portable CD player by adopting the
22 Moving Picture Experts Group (MPEG), a high compression rate algorithm, in
23 the electronic anti-shock system (EASS), so as to increase the storage capacity of
24 the temporary memory and extend the buffering time.

1 When PCM signals are received, the EASS use an MPEG compression
2 algorithm with high compression rate to convert the PCM signals to digital
3 values in the form of data streams and save them in the temporary memory.
4 Conversely, when the audio compressed data are read out from the temporary
5 memory, EASS uses the above compression algorithm to convert the audio
6 compressed data to restore to the original PCM format for sound reproduction.

7 The precondition for using the MPEG compression algorithm is that the
8 sound reproduction of the CD player shall closely resemble the sound quality in
9 the information medium (CD). Since the compression rate using MPEG is a
10 multiple of the conventional ADPCM, a double or triple amount of audio data
11 can be saved in the temporary memory, thus the buffering time of the EASS can
12 be lengthened considerably. Therefore, the system can effectively prevent
13 vibration-caused interruptions during audio playback.

14 The above audio compression format can be either MPEG 1 or MPEG 2.

15 The secondary object of the present invention is to provide an improved
16 electronic anti-shock system (EASS) for portable CD players that is capable of
17 realizing power saving. During the data operation in the buffer memory, the CD
18 servomotor can be kept in the suspended mode using minimal power, as
19 compared with the active mode, in which the servomotor becomes a major
20 power user in the system.

21 Other objectives, advantages and novel features of the invention will
22 become more apparent from the following detailed description when taken in
23 conjunction with the accompanying drawings.

24 **BRIEF DESCRIPTION OF THE DRAWINGS**

1 Fig. 1 is a comparative chart of the compression rates between MPEG
2 layer II and layer III showing data sizes in different stages;
3 Fig. 2 is a system block diagram of the present invention; and
4 Fig. 3 is a system block diagram of a conventional electronic anti-shock
5 system (EASS).

6 DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

7 The disclosed electronic anti-shock system (EASS) is created by
8 converting the input PCM signals to bit streams and saving them in a temporary
9 memory to create a data buffer and after a predetermined amount of time the
10 audio data are read out from memory and converted back to the original PCM
11 format for sound reproduction.

12 According to the present invention, a high audio compression algorithm
13 such as MPEG is used to convert the input PCM signals to audio compressed
14 data and save them in temporary memory, and after a certain time the saved
15 audio data are read out from the temporary memory and converted by the same
16 audio compression algorithm back to the original PCM format;

17 Since the compression rate tends to be inversely related to the quality of
18 sound reproduction, the adoption of a high compression rate has to be very
19 carefully considered to not affect the reproduction quality during playback. The
20 advanced compression algorithm, such as the Moving Picture Experts
21 Group(MPEG)/layer II and III, is able to produce acceptable sound quality using
22 much less memory than in an ADPCM, provided that the sound reproduction of
23 the CD player shall closely resemble the sound quality in the recording medium
24 (CD). The compression rates of MPEG Layer II, Layer III are shown in Fig. 1. It

1 is clear that the MPEG Layer III is even more powerful occupying even smaller
2 memory.

3 For a CD player having EASS, the application of Layer II and Layer III
4 will produce different results, because the buffering time for implementation
5 with Layer III will be longer than that with Layer II given the same amount of
6 memory.

7 For example, as shown in Fig. 1, using 160K bits compression rate with
8 LAYER II, and 128K bits with LAYER III; and applying this on a DRAM
9 memory with a capacity of 16M bits, the buffering time can thus be computed as
10 follows:

11 LAYER II : $16,000,000(\text{bits}) \div 160,000(\text{bps}) \approx 100 \text{ (sec)}$

12 LAYER III: $16,000,000(\text{bits}) \div 128,000(\text{bps}) \approx 125 \text{ (sec)}$

13 Compared with the ADPCM algorithm in the 3-bit and 4 bit modes, the
14 buffering time will be extended two to three times longer, thus the efficiency of
15 the EASS can be improved considerably preventing vibration-caused
16 interruptions in audio playback.

17 The structure of the present invention, as implemented in one preferred
18 embodiment is shown in Fig. 2, comprises a MPEG encoder (10), a memory
19 device (DRAM) (20), a DRAM controller (30), and an MPEG decoder (40).

20 The MPEG encoder (10) is used for converting PCM signals in the left
21 channel (s_l) and the right channel(s_r) and applying the MPEG compression
22 algorithm to produce audio compressed data streams.

23 The memory device (DRAM) (20) is used for temporarily keeping
24 audio data en route to the audio signal processor, of which the input and the

1 output are respectively connected by a FIFO buffer (21) (22), and the input
2 FIFO buffer (21) is connected to the output of the MPEG encoder (10).

3 The DRAM controller (30) is used for regulating the data flow to or
4 from the memory device (DRAM) (20), wherein the DRAM controller (30) is
5 respectively connected with the memory device (DRAM) (20) and two FIFO
6 buffers (21) (22).

7 The MPEG decoder (40) is used for decoding the audio compressed data
8 passed from the memory device (20), and restoring them to the original PCM
9 format for sound reproduction, wherein the MPEG decoder (40) is connected to
10 the memory device (DRAM) (20) through the FIFO buffer (22).

11 The above MPEG encoder (10) and MPEG decoder (40) may be in
12 compliance with either MPEG 1 or MPEG 2 specifications.

13 The data processing operation under the above mentioned architecture is
14 to be explained with reference to Fig. 2.

15 Input PCM signals of the left and right channel (s_l) (s_r) are passed to the
16 MPEG encoder (10) to produce audio compressed data, wherein the data are
17 temporarily saved in a static random access memory (SRAM) through a FIFO
18 buffer (unnumbered), which enables a dynamic configuration module to conduct
19 sideband coding, and then the audio data are further processed through
20 quantizing and packetizing to produce a digital data stream representing the
21 audio compressed data.

22 The data stream through the FIFO buffer (21) is written into the memory
23 device (DRAM) (20) by means of the DRAM controller (30). Since the MPEG
24 encoder (10) uses a high compression rate in the signal processing, the amount of

1 output data from the MPEG encoder (10) is considerably reduced, and the
2 utilization of the memory device (DRAM) (20) can thus be improved.

3 Thereafter, the audio data, saved in the memory device (20) for a
4 predetermined time, are read out and passed to the MPEG decoder (40) through
5 the FIFO buffer (22), wherein the data are first depacketized to remove the
6 encapsulation over the data, and then further through reverse quantizing and
7 phase negation to restore to the original PCM format. After further
8 reconfiguration and signal processing, the audio signals are played back over the
9 speaker.

10 From the foregoing, it is apparent that the instrumentality of the present
11 invention is to increase the compression rate of the EASS so as to increase the
12 utilization of temporary memory. As a result, the system can prevent vibration or
13 shock-caused interruptions during audio playback. However, the precondition to
14 using the high compression rate algorithm is that the quality of sound
15 reproduction of the CD player has to closely resemble the original recording
16 level on the information medium (CD). If the above condition can be satisfied,
17 the buffering time can be effectively lengthened, and the power saving can also
18 be realized by increasing the time of the CD servomotor in the suspended mode.

19 When compared with the ADPCM compression algorithm, double or
20 triple amounts of audio data can be saved in the same amount of memory, and
21 the buffering time of the EASS can be lengthened considerably.

22 According to the present invention, the CD player having EASS is able
23 to achieve the most desirable balance point between audio performance, power
24 saving and low cost.

1 It is to be understood, however, that even though numerous
2 characteristics and advantages of the present invention have been set forth in the
3 foregoing description, together with details of the structure and function of the
4 invention, the disclosure is illustrative only, and changes may be made in detail,
5 especially in matters of shape, size, and arrangement of parts within the
6 principles of the invention to the full extent indicated by the broad general
7 meaning of the terms in which the appended claims are expressed.